Asterisk introductie en workshop

New face, who this?

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Na deze workshop weet je ...

- Hoe SIP in de basis werkt
- Wat Asterisk is
- Hoe je een SIP extensie registreert en gebruikt
- Hoe je verbinding maakt met een provider via een SIP trunk
- Hoe je basis VoIP problemen opspoort en oplost
- Hoe je Asterisk kan configureren om te bellen en gebeld te kunnen worden
- Wat je nodig hebt om zelf verder te gaan

1 Voorkennis

VoIP, hoe werkt dat eigenlijk?

SIP, SDP, RTP

en andere DLA's

- Session Initiation Protocol
- Session Description Protocol
- Real-Time Transport Protocol
- Real-Time Transport Control Protocol
- PSTN

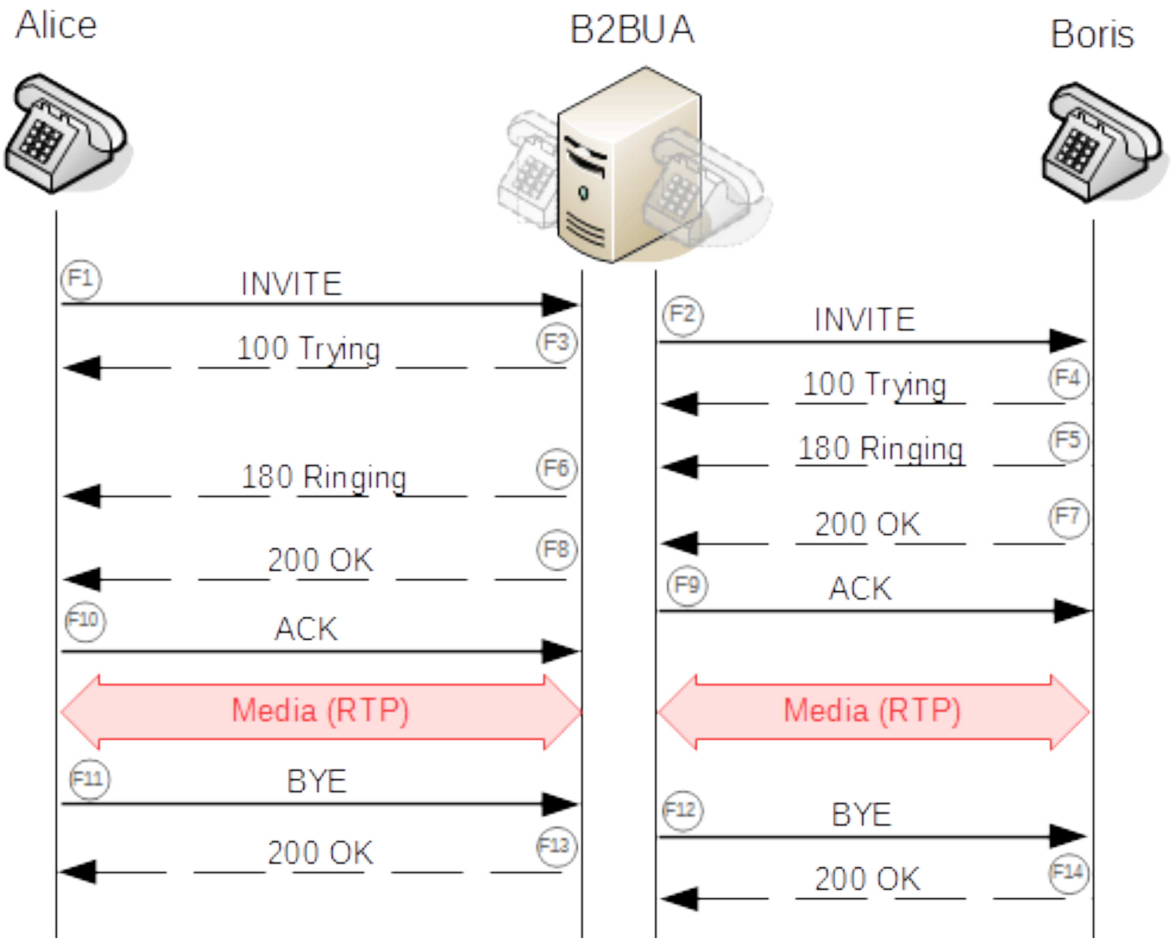
Meer terminologie

- SIP endpoints
- SIP trunks
- Extensions
- Dialplans

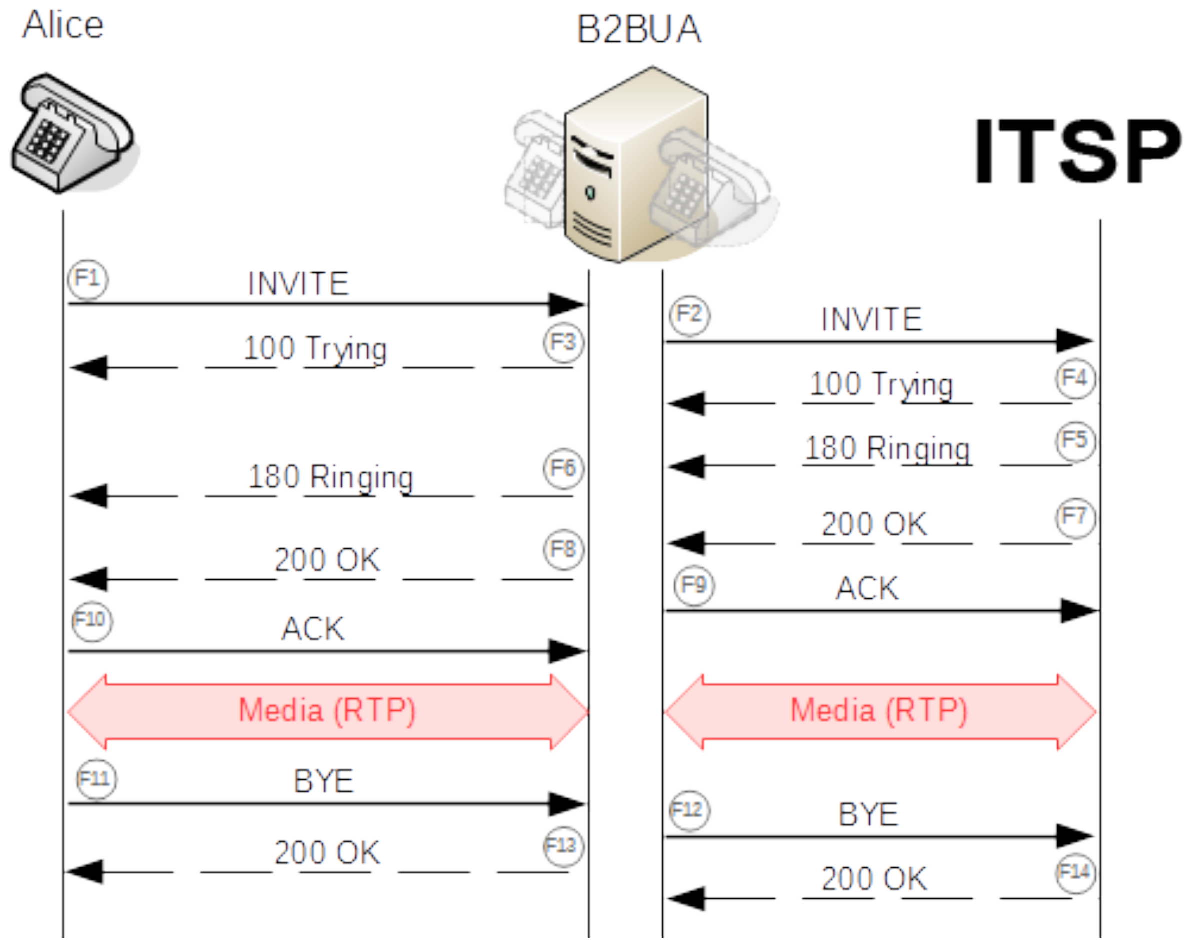
SIP methods

- INVITE - initiates a session
- ACK - confirms a request
- BYE - terminates a session
- CANCEL - cancels a pending INVITE
- OPTIONS - capability inquiry
- REGISTER - binds an address to location
- SUBSCRIBE - subscribe to events from a notifier
- NOTIFY - notify the subscriber

SIP **flows**


Alice
B2BUA
Boris
F1
INVITE
F2
INVITE
F3
100 Trying
F4
100 Trying
F5
180 Ringing
F6
180 Ringing
F7
200 OK
F8
200 OK
F9
ACK
F10
ACK
Media (RTP)
Media (RTP)
F11
BYE
F12
BYE
F13
200 OK
F14
200 OK

SIP flows

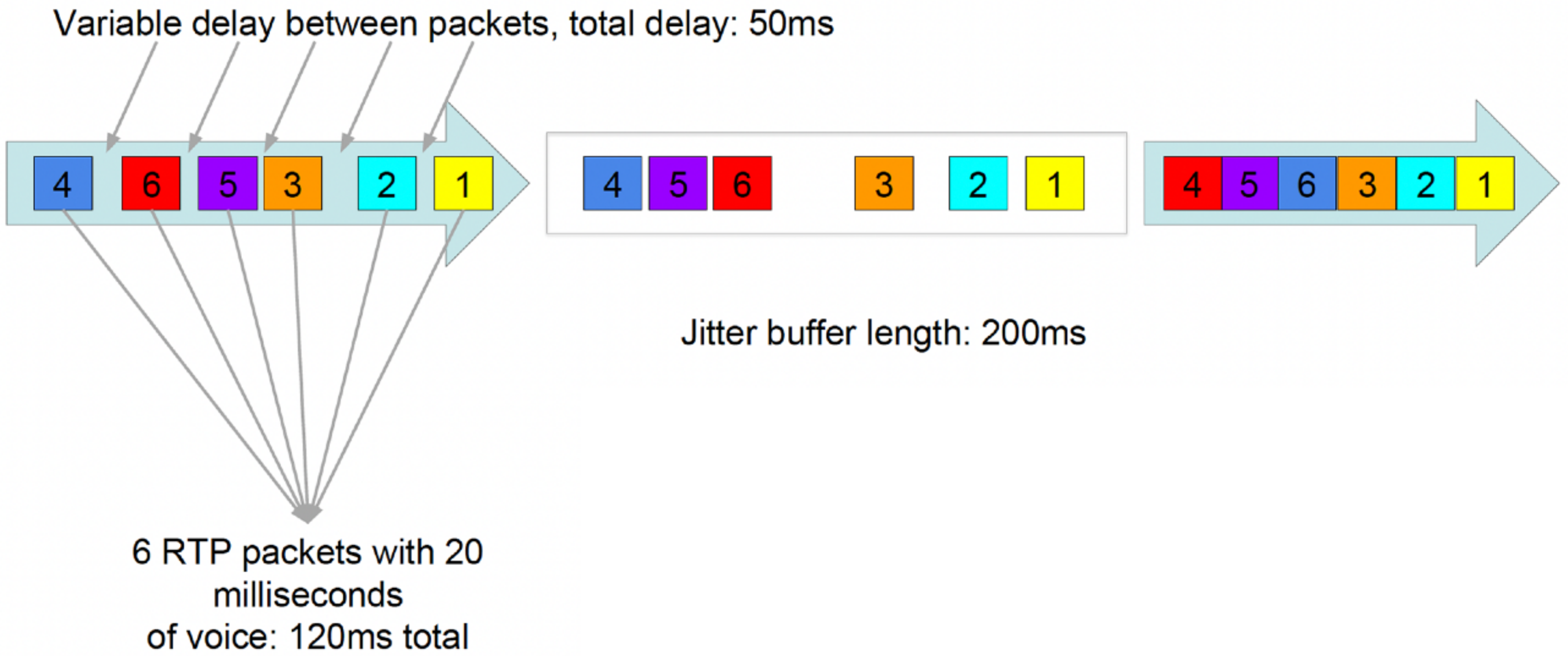

Alice
B2BUA
ITSP
F1
INVITE
F2
INVITE
F3
100 Trying
F4
100 Trying
F5
180 Ringing
F6
180 Ringing
F7
200 OK
F8
200 OK
F9
ACK
F10
ACK
Media (RTP)
Media (RTP)
F11
BYE
F12
BYE
F13
200 OK
F14
200 OK

Codecs

Can you hear me now?

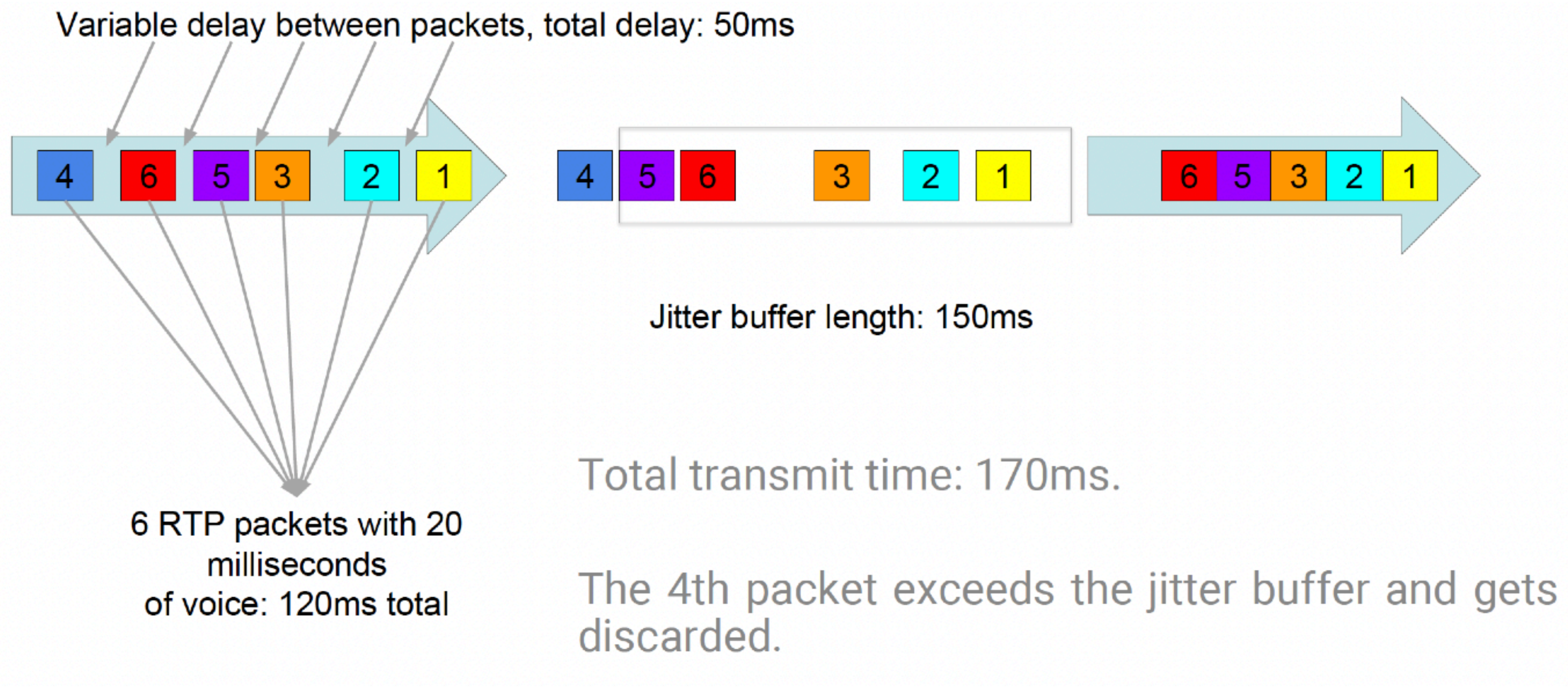
Codec Information				Bandwidth Calculations			
Codec & Bit Rate (Kbps)	**Codec Sample Size (Bytes)**	**Codec Sample Interval (ms)**	**Mean Opinion Score (MOS)**	**Voice Payload Size (Bytes)**	**Voice Payload Size (ms)**	**Packets Per Second (PPS)**	**Bandwidth Ethernet (Kbps)**
G.711 (64 Kbps)	80 Bytes	10 ms	4.1	160 Bytes	20 ms	50	87.2 Kbps
G.729 (8 Kbps)	10 Bytes	10 ms	3.92	20 Bytes	20 ms	50	31.2 Kbps
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	3.9	24 Bytes	30 ms	33.3	21.9 Kbps
G.723.1 (5.3 Kbps)	20 Bytes	30 ms	3.8	20 Bytes	30 ms	33.3	20.8 Kbps
G.726 (32 Kbps)	20 Bytes	5 ms	3.85	80 Bytes	20 ms	50	55.2 Kbps
G.726 (24 Kbps)	15 Bytes	5 ms		60 Bytes	20 ms	50	47.2 Kbps
G.728 (16 Kbps)	10 Bytes	5 ms	3.61	60 Bytes	30 ms	33.3	31.5 Kbps
G722_64k(64 Kbps)	80 Bytes	10 ms	4.13	160 Bytes	20 ms	82.8 Kbps	87.2 Kbps
ilbc_mode_20(15.2Kbps)	38 Bytes	20 ms	NA	38 Bytes	20 ms	50	38.4Kbps
ilbc_mode_30(13.33Kbps)	50 Bytes	30 ms	NA	50 Bytes	30 ms	33.3	28.8 Kbps

VoIP challenges
Jitter



VoIP challenges

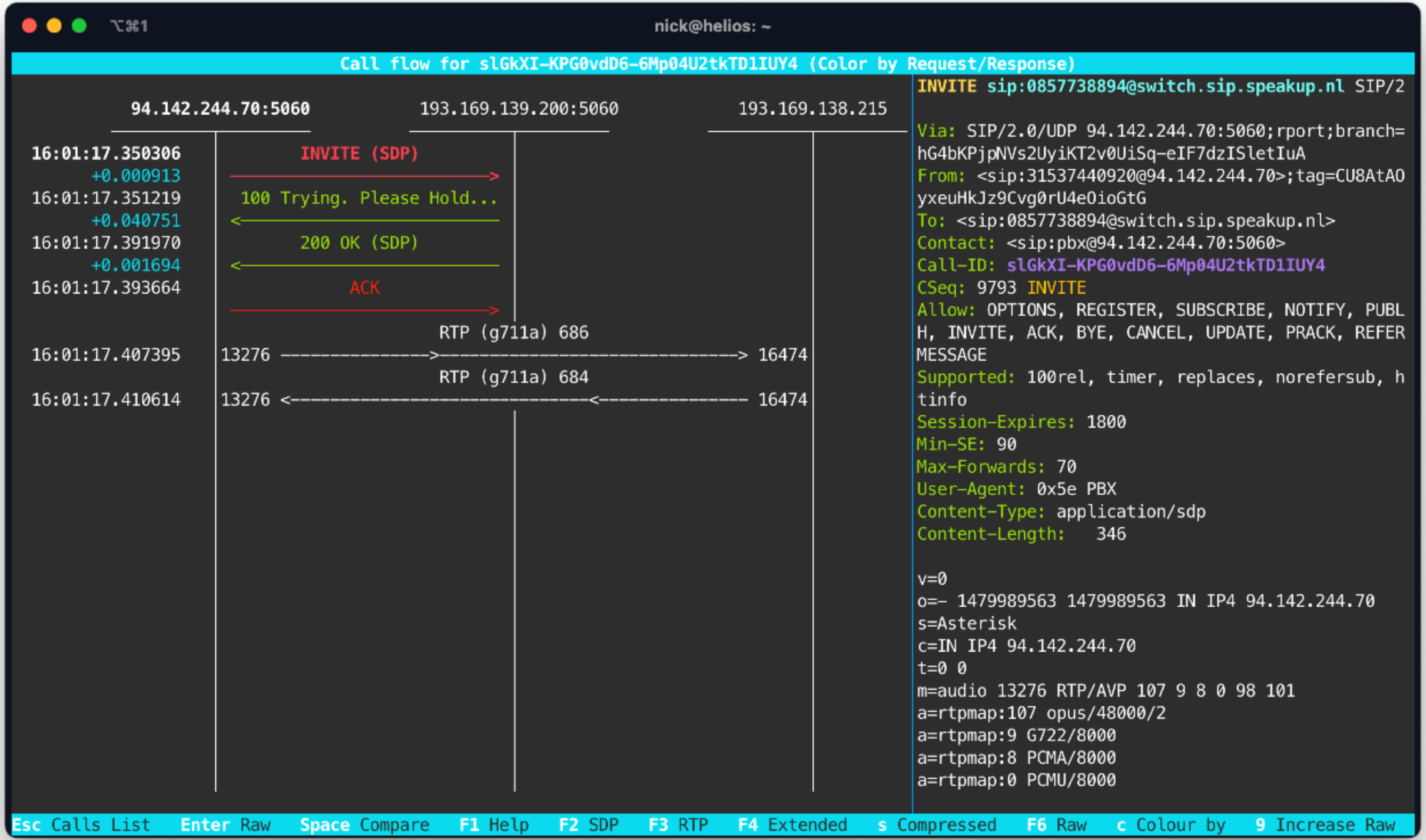
Jitter! *Aaargh*

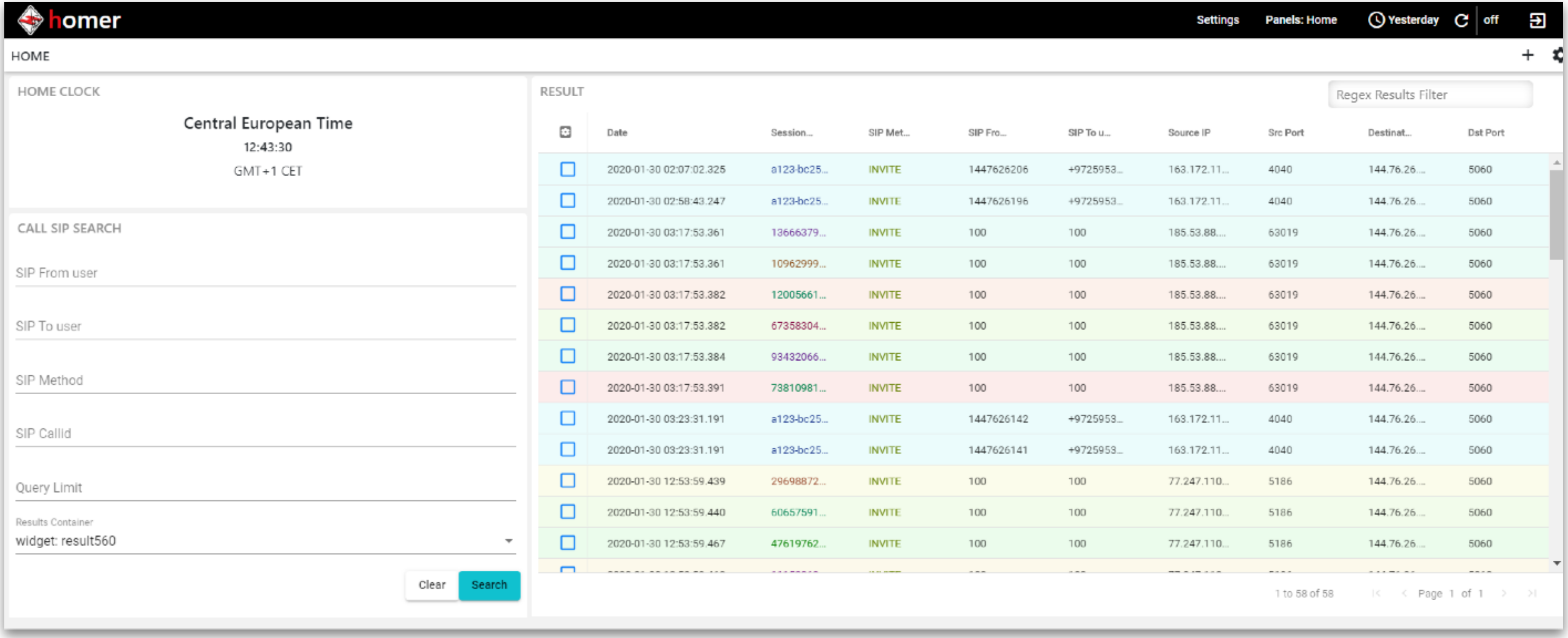


VoIP challenges

Latency / Delay

- Een stabiele netwerkomgeving is essentieel
- Lagere ping is meer beter
- One-way audio? Problemen met signalering?
  - Check je firewall
  - Schakel je ALG uit


nick@helios: ~
Call flow for slGkXI-KPG0vdD6-6Mp04U2tkTD1IUY4 (Color by Request/Response)
94.142.244.70:5060
193.169.139.200:5060
193.169.138.215
16:01:17.350306
INVITE (SDP)
+0.000913
16:01:17.351219
100 Trying. Please Hold...
+0.040751
16:01:17.391970
200 OK (SDP)
+0.001694
16:01:17.393664
ACK
RTP (g711a) 686
16:01:17.407395
13276 --------------->----------------------------> 16474
RTP (g711a) 684
16:01:17.410614
13276 <-------------------------------<--------------- 16474
INVITE sip:0857738894@switch.sip.speakup.nl SIP/2
Via: SIP/2.0/UDP 94.142.244.70:5060;rport;branch=
hG4bKPjpNVs2UyiKT2v0UiSq-eIF7dzISletIuA
From: <sip:31537440920@94.142.244.70>;tag=CU8AtAO
yxeuHkJz9Cvg0rU4eOioGtG
To: <sip:0857738894@switch.sip.speakup.nl>
Contact: <sip:pbx@94.142.244.70:5060>
Call-ID: slGkXI-KPG0vdD6-6Mp04U2tkTD1IUY4
CSeq: 9793 INVITE
Allow: OPTIONS, REGISTER, SUBSCRIBE, NOTIFY, PUBL
H, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, REFER
MESSAGE
Supported: 100rel, timer, replaces, norefersub, h
tinfo
Session-Expires: 1800
Min-SE: 90
Max-Forwards: 70
User-Agent: 0x5e PBX
Content-Type: application/sdp
Content-Length: 346
v=0
o=- 1479989563 1479989563 IN IP4 94.142.244.70
s=Asterisk
c=IN IP4 94.142.244.70
t=0 0
m=audio 13276 RTP/AVP 107 9 8 0 98 101
a=rtpmap:107 opus/48000/2
a=rtpmap:9 G722/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
Esc Calls List
Enter Raw
Space Compare
F1 Help
F2 SDP
F3 RTP
F4 Extended
s Compressed
F6 Raw
c Colour by
9 Increase Raw


homer
Settings
Panels: Home
Yesterday
off
HOME
HOME CLOCK
Central European Time
12:43:30
GMT+1 CET
CALL SIP SEARCH
SIP From user
SIP To user
SIP Method
SIP CallId
Query Limit
Results Container
widget: result560
Clear
Search
RESULT
Regex Results Filter
Date
Session...
SIP Met...
SIP Fro...
SIP To u...
Source IP
Src Port
Destinat...
Dst Port
2020-01-30 02:07:02.325 a123-bc25... INVITE 1447626206 +9725953... 163.172.11... 4040 144.76.26... 5060
2020-01-30 02:58:43.247 a123-bc25... INVITE 1447626196 +9725953... 163.172.11... 4040 144.76.26... 5060
2020-01-30 03:17:53.361 13666379... INVITE 100 100 185.53.88.... 63019 144.76.26... 5060
2020-01-30 03:17:53.361 10962999... INVITE 100 100 185.53.88.... 63019 144.76.26... 5060
2020-01-30 03:17:53.382 12005661... INVITE 100 100 185.53.88.... 63019 144.76.26... 5060
2020-01-30 03:17:53.382 67358304... INVITE 100 100 185.53.88.... 63019 144.76.26... 5060
2020-01-30 03:17:53.384 93432066... INVITE 100 100 185.53.88.... 63019 144.76.26... 5060
2020-01-30 03:17:53.391 73810981... INVITE 100 100 185.53.88.... 63019 144.76.26... 5060
2020-01-30 03:23:31.191 a123-bc25... INVITE 1447626142 +9725953... 163.172.11... 4040 144.76.26... 5060
2020-01-30 03:23:31.191 a123-bc25... INVITE 1447626141 +9725953... 163.172.11... 4040 144.76.26... 5060
2020-01-30 12:53:59.439 29698872... INVITE 100 100 77.247.110... 5186 144.76.26... 5060
2020-01-30 12:53:59.440 60657591... INVITE 100 100 77.247.110... 5186 144.76.26... 5060
2020-01-30 12:53:59.467 47619762... INVITE 100 100 77.247.110... 5186 144.76.26... 5060
1 to 58 of 58
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2 Asterisk

Het Zwitserse zakmes van de telefonie

Wat is Asterisk

- PBX, VoIP gateway,
- B2BUA
- Communicatie framework

Hoe configureer je Asterisk

- Config files
- Asterisk REST interface
- Asterisk Management Interface
- Database

swagger | http://localhost:8088/ari/api-docs/resources.json | asterisk:asterisk | Explore

asterisk : Asterisk resources — Show/Hide | List Operations | Expand Operations | Raw

endpoints : Endpoint resources — Show/Hide | List Operations | Expand Operations | Raw

channels : Channel resources — Show/Hide | List Operations | Expand Operations | Raw

GET	/channels	List all active channels in Asterisk.
POST	/channels	Create a new channel (originate).
POST	/channels/create	Create channel.
GET	/channels/{channelId}	Channel details.
POST	/channels/{channelId}	Create a new channel (originate with id).
DELETE	/channels/{channelId}	Delete (i.e. hangup) a channel.
POST	/channels/{channelId}/continue	Exit application; continue execution in the dialplan.
POST	/channels/{channelId}/move	Move the channel from one Stasis application to another.
POST	/channels/{channelId}/redirect	Redirect the channel to a different location.
POST	/channels/{channelId}/answer	Answer a channel.
POST	/channels/{channelId}/ring	Indicate ringing to a channel.
DELETE	/channels/{channelId}/ring	Stop ringing indication on a channel if locally generated.
POST	/channels/{channelId}/dtmf	Send provided DTMF to a given channel.
POST	/channels/{channelId}/mute	Mute a channel.
DELETE	/channels/{channelId}/mute	Unmute a channel.
POST	/channels/{channelId}/hold	Hold a channel.

Hoe configureer je Asterisk

- pjsip.conf
  - SIP endpoints
- extensions.conf
  - Dialplans
- modules.conf
- rtp.conf

```
[transport-udp]
type=transport
protocol=udp
bind=0.0.0.0

[6001]
type=endpoint
context=from-internal
disallow=all
allow=ulaw
auth=6001
aors=6001

[6001]
type=auth
auth_type=userpass
password=unsecurepassword
username=6001

[6001]
type=aor
max_contacts=1
```

```
[from-internal]
exten = 200,1,Answer()
same = n,Wait(1)
same = n,Playback(hello-world)
same = n,Hangup()
```

```
[from-internal]
exten = 200,1,Answer()
same = n,Wait(1)
same = n,Playback(hello-world)
same = n,Hangup()

exten = _X.,1,Noop(Outgoing call to ${EXTEN} via Speakup)
same.= n,Dial(PJSIP/${EXTEN}@speakup)
same = n,Hangup()

[from-speakup]
exten = 31887732587,1,Noop(Incoming call on Speakup trunk from ${CALLERID(num)})
same = n,Dial(PJSIP/200)
same = n,Hangup()
```

Safety third

- SIP scanners
- Fraude
- sipvicious en de 'friendly-scanner'

Let's get our hands dirty!

Tijd om aan de slag te gaan

Stap 1

- Maak een verbinding met de softphone op je pc
    - Pro tip: Voeg je IP-adres toe aan de de firewall. Er staat `ufw` op de VPS.
    - `$ ufw allow from [ip adres]`
- Maak extensie 200 aan in je dialplan en zorg ervoor dat Asterisk dit gesprek beantwoord met een geluidsbestand
    - bijvoorbeeld 'hello-world' of 'tt-weasels'

```
[transport-udp]
type=transport
protocol=udp
bind=0.0.0.0

[6001]
type=endpoint
context=from-internal
disallow=all
allow=ulaw
auth=6001
aors=6001

[6001]
type=auth
auth_type=userpass
password=unsecurepassword
username=6001

[6001]
type=aor
max_contacts=1
```

Stap 2

- Maak een verbinding met de Speakup trunk waarvan je de credentials al hebt gekregen
- Zorg dat je publieke telefoonnummer bereikbaar wordt.
  - Om te testen kan je een geluidsbestand configureren, maar zorg dat het gesprek op je softphone uit komt.

https://github.com/speakupnl/asterisk-configuration

Stap 3

- Zorg dat je ook uitgaand kunt bellen
- Gebruik bijvoorbeeld een Speakup testlijn om te kijken of je kunt bellen. Uiteraard mag je ook je eigen 06-nummer bellen.
- Of bel elkaar? :)

- Echo: 085 7738891
- Muziek: 085 7738894

```
[from-internal]
exten = 200,1,Answer()
same = n,Wait(1)
same = n,Playback(hello-world)
same = n,Hangup()

exten = _X.,1,Noop(Outgoing call to ${EXTEN} via Speakup)
same.= n,Dial(PJSIP/${EXTEN}@speakup)
same = n,Hangup()

[from-speakup]
exten = 31887732587,1,Noop(Incoming call on Speakup trunk from ${CALLERID(num)})
same = n,Dial(PJSIP/200)
same = n,Hangup()
```

3

Wrapping up

Where to take it from here?

Next steps

- Telemarketer torture
  - Je eigen AaaS?
- Nummerherkenning voor familie / vrienden
- Voicemail
- Bellers identificeren met Google Places API, data van de ACM of uit andere bronnen …
- Zork over IP
- Project MF

Dank je wel!

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